Chapter 9

HSPA Transport Network Layer Congestion Control

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9.1 Introduction

The introduction of High-Speed Packet Access (HSPA) greatly improves the achievable bit rate but it presents new challenges to be solved in the Wideband Code Division Multiple Access (WCDMA) radio access network (RAN). For dedicated channels (DCHs), transport network bandwidth can be reserved by means of admission control. Bandwidth reservation is not efficient for HSDPA because of the higher peak rates and much higher variance of achieved bit rate; thus a new solution is needed to control congestion. In the Internet, such congestion is controlled by the end-user Transmission Control Protocol (TCP), but that is not possible in the access transport network because lost packets are retransmitted by WCDMA-specific lower layers.

This chapter describes the Iub/Iur transport network and congestion control solutions employed to efficiently utilize the transport network bottleneck. The main focus of the chapter is HSPA flow control, but it also describes other congestion avoidance mechanisms, namely buffering and quality-of-service differentiation, call admission control, and link dimensioning. It also provides insight into how the different congestion avoidance mechanisms cooperate. The architecture of flow control, the congestion detection mechanisms, and the possible congestion actions are described. Different flow control solutions are compared and a few case studies are presented. Finally, the HSPA Framing Protocol (FP), which provides the HSPA flow control framework, is described in detail and transport network overhead is evaluated.

9.1.1 Iub/Iur Transport Network Architecture

Figure 9.1 gives a schematic view of a WCDMA network, which consists of user equipment (UE), the WCDMA terrestrial radio access network (WCDMA RAN), and the core network. WCDMA RAN (WRAN) is also called Universal Mobile Telecommunications System (UMTS) Terrestrial Radio Access Network (UTRAN).

The WCDMA RAN handles all tasks that relate to radio access control, such as radio resource management and handover control. The core network, which is the backbone of WCDMA, connects the access network to external networks (e.g., PSTN, Internet). The user equipment (mobile

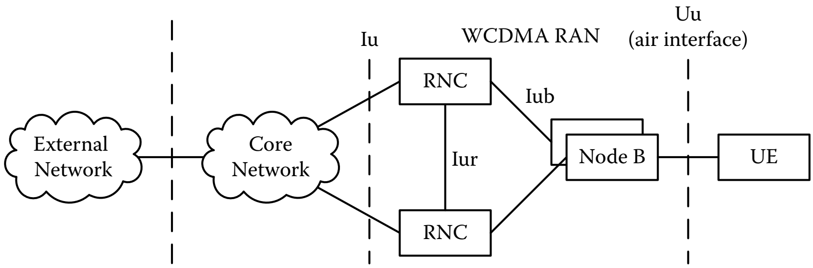

Figure 9.1 Schematic view of a WCDMA network.

terminal or mobile station) is connected to radio base stations (Node B) over the WCDMA air interface (Uu). During soft handover, one UE can communicate with several Node Bs simultaneously.

According to the WCDMA RAN specifications by the 3rd Generation Partnership Project (3GPP), all radio network functions and protocols are separated from the functions and protocols in the transport network layer (TNL). The WCDMA RAN transport network transmits data and control information between the radio network controller (RNC) and the Node Bs (Iub interface) or between RNCs (Iur interface). For a given UE there is always a single serving RNC (SRNC), which terminates the user and control plane protocols of that UE. In this chapter whenever we write RNC, we mean this SRNC. The transport network layer provides data and signaling bearers for the radio network application protocols between RAN nodes, and includes transport network control-plane functions for establishing and releasing such bearers when instructed to do so by the radio network layer.

The first release of the 3GPP specification was called Release 99. Figure 9.2 shows the Release 99 protocol stack at the Iub interface for transferring data streams on common transport channels (CCHs) and dedicated transport channels (DCHs) to the air interface.

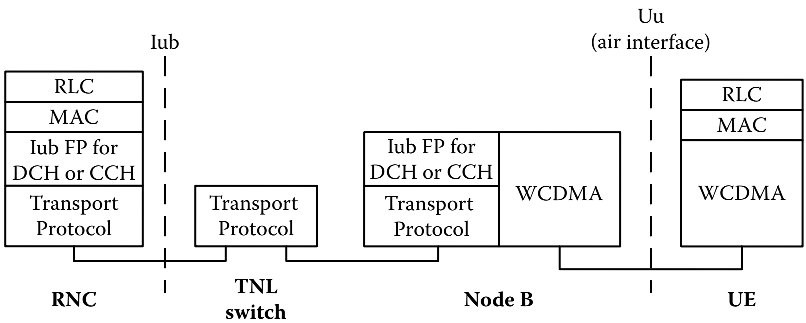

Figure 9.2 User plane protocol stack between RNC and Node B for Release 99.

The acknowledged retransmission mechanism of the radio link control (RLC) protocol ensures reliable transmission of loss-sensitive traffic over the air interface. RLC provides three types of service to the upper layer according to the standard [1]: transparent (no additional protocol information), unacknowledged (delivery not guaranteed), and acknowledged data transfer mode. The normal RLC mode for packet-type services is the acknowledged mode, so RLC works in this mode in the case of HSPA as well. The RLC protocol is used by both signaling radio bearers and radio bearers for packet-switched data services, but not by radio bearers for circuit-switched services. RLC Acknowledged Mode (AM) [1], which is a selective repeat automatic repeat request (SR-ARQ) protocol, provides transport service to upper layers between UE and the RNC. RLC AM does not include congestion control functionality because it assumes that RLC protocol data units (PDUs) are transmitted by the Medium Access Control–dedicated (MAC-d) layer according to the available capacity allocated on Uu and the transport network. The RLC status messages, which are being sent regularly based on preset events, trigger retransmission of all missing PDUs. The most dominant of these terms and events in the sending time interval. The receiver side detects missing RLC PDUs based on gaps in the sequence numbers. The receiver side RLC requests retransmission by sending back a status PDU, informing which PDUs within the receiving window have been acknowledged (ACK) or lost (negative ACK, NACK). Upon reception of a status message, the sender can slide its transmission window (Tx window) if one or more in-sequence frames are acknowledged, so that new PDUs can be sent. If there are NACKs in the status message, the sender retransmits the missing PDUs, giving them priority over new ones. Several unsuccessful retransmissions of the same PDU trigger an RLC reset and the whole RLC Tx window (maximum ~80 kBytes in the case of 42-Byte RLC PDUs) is discarded.

In Release 99, the MAC-d protocol forms sets of transport blocks in the air interface and schedules them according to the timing requirements of WCDMA. Each scheduled period, called a transmission time interval (TTI), is 10 ms in length or multiples thereof.

Release 99 WCDMA radio connections, or radio access bearers (RABs), have practical bit rate values between 8 and 384 kbps. The size of the MAC transport blocks and the length of the TTI are RAB specific.

For data transfer over the Iub interface, the MAC transport blocks are encapsulated into Iub frames according to the Iub user-plane (UP) protocol for CCH or DCH data streams. Each Iub UP data stream needs a separate transport network connection between the RNC and Node B. The transport network thus establishes one TNL connection for each data stream. In the TNL switch (optional) is used for building aggregating transport networks.

In response to the increased need for higher bit rate and more efficient transmission of packet data over cellular networks, WCDMA 3GPP Release 5 extended the WCDMA specification with High-Speed Downlink Packet

Access (HSDPA) [2]. The demand for uplink performance improvement was addressed by introducing Enhanced Dedicated Channel (E-DCH)—often referred as Enhanced Uplink (EUL) or High-Speed Uplink Packet Access (HSUPA)—in 3GPP Release 6 [3]. HSDPA and HSUPA together are called HSPA. The main architectural novelty of HSPA is that certain parts of the control of radio resources have been moved from RNC to Node Bs. In 3GPP Release 7, higher-order modulation and multiple-input, multiple-output (MIMO) are introduced for HSDPA to further improve the achievable bit rate [4].

For HSDPA, a new shared downlink transport channel, called High-Speed Downlink Shared Channel (HS-DSCH), is also introduced. This channel is dynamically shared among packet data users, primarily in the time domain; the TTI is 2 ms. The application of shared channel makes the use of available radio resources in WCDMA more efficient. HSDPA also supports new features that rely on the rapid adaptation of transmission parameters to instantaneous radio conditions. The main principles are fast link adaptation, fast Hybrid ARQ (HARQ) with soft-combining, and fast channel-dependent scheduling [5]. To support HSDPA with minimum impact on the existing radio interface protocol architecture, a new Medium Access Control (MAC) sublayer, MAC-hs, has been introduced for HS-DSCH transmission and is implemented in Node B and UE. In Release 7, higher-order modulation and MIMO are introduced. To support the higher bit rates achievable, layer-2 enhancements became necessary. Previously allowed RLC PDU sizes do not support these bit rates; therefore, RLC PDU size must be increased. However, increasing the fixed RLC PDU size would result in higher padding in RLC and over the air interface, and the increased fixed size would also result in smaller cell coverage, due to the larger minimum air interface transport format. Therefore, flexible RLC PDU sizes were introduced and an improved MAC layer, MAC-ehs, introduced segmentation and multiplexing [4].

For HSUPA, new MAC layers (MAC-e/-es) were introduced to support the new features, that is, fast Hybrid Automatic Repeat Request (HARQ) with soft combining, reduced TTI (2 ms), and fast scheduling. HSUPA was further improved with the possibility of higher-order modulation in Release 7 [6]. The Release 6 and 7 improvements allow layer-1 peak rates up to 5.7 Mbps and 11 Mbps in uplink, respectively.

Despite the fact that similar features have been introduced for HSDPA and HSUPA, there are several essential differences [3]. In the case of HSDPA, the HS-DSCH is shared in the time domain among all users; for HSUPA, the E-DCH is dedicated to a user. For HSDPA, the transmission power is kept more or less fixed and rate adaptation is used. However, this is not possible for HSUPA because the uplink is non-orthogonal; therefore, fast power control is needed for fast link adaption. Consequently, for HSDPA, the shared resources are the transmission power and the code space of the shared channel; but for HSUPA, the shared resource is the interference

headroom. Soft handover is not supported by HSDPA, while for HSUPA, soft handover is used to decrease the interference from neighboring cells and to have macro diversity gain.

The main architectural novelty of HSPA is that the control of radio frame scheduling has been moved from RNC to Node Bs. While fixed capacity (e.g., 64 kbps) can be reserved for traditional DCH traffic in the access transport network, per-flow bandwidth reservation is not efficient for HSPA because air interface throughput is much higher and fluctuates more. If bandwidth reservation is not used, then congestion can happen in both the Iub transport network and the air interface. In the current architecture, TCP cannot efficiently resolve a congestion situation in the access network because RLC AM retransmissions between the RNC and the UE hide the congestion situations from TCP. Thus, a flow control function has been introduced to control the data transfer between the RNC and Node B. Originally, HSDPA flow control was designed to take only the transmission capabilities of the air interface into account and to limit the RLC round-trip time (RTT) and the number of RLC stalls [7]. However, in practice, the increased air interface capacity did not always come with similarly increased Iub transport network capacity. Network operators often upgrade the Node Bs first and delay the upgrade of the transport network until there is significant HSPA traffic. In some cases also, the cost of Iub transport links is still high; however, it decreases significantly with the introduction of new mobile backhaul technologies [8]. Thus it is a common scenario that the throughput is limited by the capacity available on the Iub transport network links and not by the capacity of the air interface. On these bottleneck transport network links, it is important to maintain high efficiency. For HSDPA, it has been identified in 3GPP that the flow control framework can also resolve these congestion situations if a transport network congestion detection functionality is available. The design of the HSUPA framing protocol already took this aspect into account. For this purpose, congestion detection-specific fields to the Iub HS-DSCH Data Frame [7,9–11] and a new Information Element for the Iub/Iur Framing Protocol E-DCH Data Frame were introduced [12]. The requirements and principles of HSDPA and HSUPA congestion control are summarized in [13].

TCP slow start normally increases the TCP congestion window (cwnd) size fast to its maximum when a new flow arrives it is kept at that value, because due to RLC retransmissions, TCP does not experience loss (except, for example, RLC reset). In case of transport network congestion, RLC keeps retransmitting lost PDUs until they are successfully received (or until RLC resets); however, these retransmissions even increase the congestion level. In this way, due to the aforementioned reasons, TCP is notified significantly later about the transport network congestion. As a summary, we need a system-specific congestion control (CC) because the RLC does not have congestion control functionality and TCP congestion control cannot operate

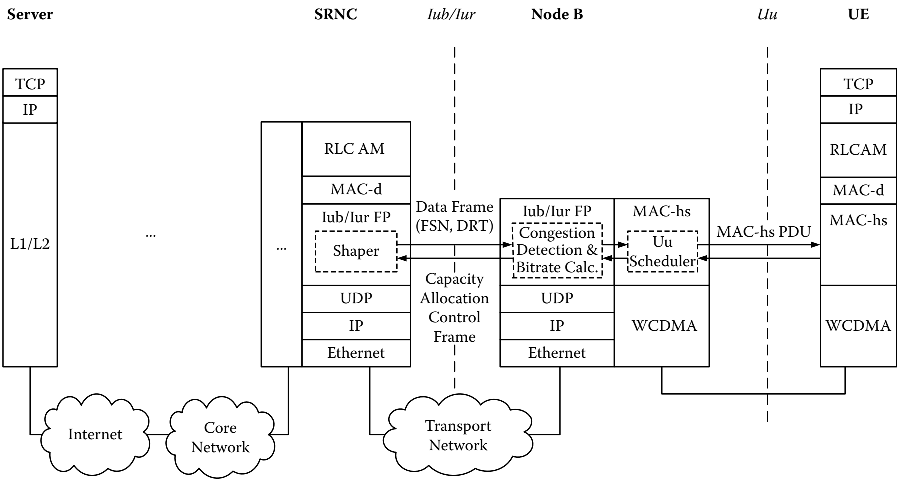

Figure 9.3 HSDPA protocol stack and flow control architecture.

efficiently above the RLC AM protocol. The locations of RLC AM and HARQ are depicted on Figure 9.5. Note that TCP is able to control congestion on the interfaces, which are outside the RLC retransmission. Such interfaces include the Iu interface in WCDMA and the S1 interface in the case of 3GPP Long-Term Evolution (LTE).

The nodes and protocol layers involved in the HSDPA and HSUPA flow control (FC) are depicted in Figure 9.3 and Figure 9.4, respectively [14]. The figures also show the location of the flow control-related functionalities in boxes with dashed lines.

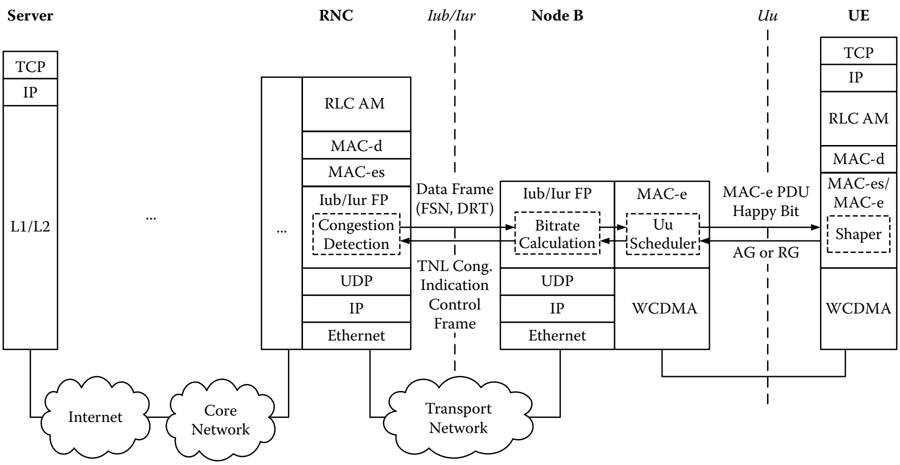

Figure 9.4 HSUPA protocol stack and flow control architecture.

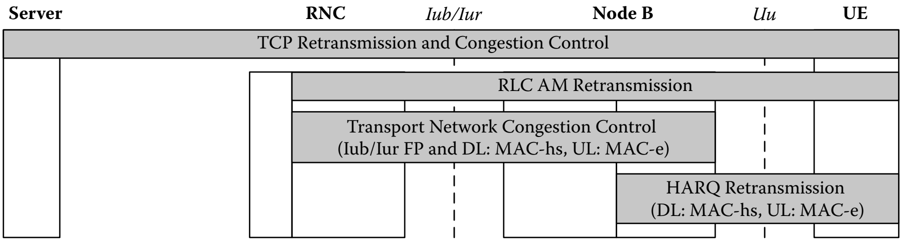

Figure 9.5 Protocol layers performing congestion control and/or retransmission.

The task of the HSDPA flow control is to regulate the transfer of MAC-d PDUs from the RNC to Node B. And the task of HSUPA flow control is to regulate the transfer of MAC-es PDUs on the Iub/Iur transport network toward the RNC. Several of these flows can share the same air interface (e.g., a cell) or transport network bottleneck (e.g., a link).

Note that for HSUPA, the regulation provided by FC is needed only when the Transport Network limits the performance. When the transport network is not limiting, the FC has no effect on the flows.

Figure 9.5 depicts the protocol layers that perform congestion control and/or retransmission. Over the air interface, fast Hybrid-ARQ is used to correct transmission errors. This results in a very small residual error probability on the air interface. Any transport network congestion is resolved by transport network congestion control and the retransmission of packets lost on transport network is done by RLC AM. TCP is used for congestion control and retransmission between the server and the UE.

9.1.2 Requirements, Different Traffic Types

The WCDMA RAN transport layer services must meet stringent quality-of-service (QoS) requirements. The most important measure of QoS performance on the Iub and Iur interfaces is maximum packet delay [15].

When HSPA is carrying moderate-speed QoS-sensitive traffic, QoS can be guaranteed by transport network bandwidth reservation by means of transport network admission control; flow control is not used. For best-effort (BE) traffic, bandwidth reservation is not efficient and flow control is used instead. HSPA carries BE traffic but HSPA Iub requires better than BE transport network connection because part of the control of radio resources is in RNC. When QoS-sensitive and BE traffic coexist in a system, the QoS-sensitive traffic is usually prioritized over BE traffic, and the capacity not used by the QoS-sensitive traffic can be utilized by the BE traffic.

9.1.3 Transport Network Solutions

With the "Iub cloud" logic [13], the traffic injected by the sending node (e.g., the RNC in the case of HSDPA) is less tightly controlled; that is, it is likely to inject too much traffic into the network, thus yielding a congestion situation. This approach should allow for statistical multiplexing in some scenarios without complex configuration of TNL topology in the RNC. However, this requires congestion detection in the receiving node (e.g., Node B in case of HSDPA) for congestion control. This is assumed to be the mainstream solution in this chapter because it supports a general transport network.

Using "Iub pipe" logic [13], the sending node enforces the traffic limit injected on the Iub interface, so it is able to instantaneously detect any congestion situation. The advantage of this approach is that there is no need to use congestion detection mechanisms in the receiving node because congestion detection is instantaneous—the only place it can occur is at the "pipe" entry. The drawback of "pipe" logic is that in some scenarios it may require complex configuration of TNL topologies in the sending node in order to leverage statistical multiplexing.

AAL2/ATM (ATM Adaptation Layer 2/Asynchronous Transfer Mode) or UDP/IP (User Datagram Protocol/Internet Protocol) is used as a transport protocol; in Figures 9.3 and 9.4, the UDP/IP/Ethernet solution is depicted as an example.

The initial WCDMA RAN specifications specified that the transport network layer must be based on ATM and AAL2 technology. However, Release 5 of the 3GPP specification also includes the option of an IP-based transport network.

For the AAL2/ATM transport solution, the data frames (DFs) are segmented into AAL2 common part sublayer (CPS) PDUs. These CPS PDUs are then fit into one or two ATM cells. There is no early packet discard for AAL2 queues; consequently, the end of the data frames can be lost, while the beginning of the data frame is still using the transport network capacity in vain because they will be discarded in the Node B anyway. We call these frames as *destroyed frames*. This behavior can be disadvantageous in the case of system overload. A detailed description of AAL2/ATM can be found in [16]. In the case of UDP/IP, the DFs to be transmitted can be larger than the maximum transfer unit (MTU) of the system, especially in case of high throughput. In this case, IP fragmentation is needed and DFs might be destroyed, as in case of AAL2/ATM. In most cases, however, the size of the DF is smaller than the MTU, and then a DF is either completely lost or transmitted. In the case of UDP/IP, the most commonly used is the layer-2 protocol Ethernet, but other layer-2 protocols are possible (e.g., MLPPP, Multi-Link Point-to-Point Protocol).

9.1.3.1 Typical Capacities and Architecture

For both HSDPA [17] and HSUPA, the Iub and Iur transport network links could be a bottleneck in the RAN. The transport network links are expected to be bottleneck e.g. in the case of E1, T1 transmissions[1] and ADSL transmission. For higher notes, e.g., E3 transmission[1] or 100 Mbps Ethernet access, the TN bit rate is higher than the typical achievable air interface throughput; however, in case of very good air interface conditions, TN can still be a bottleneck. In most networks, it is expected that in a significant percentage of the cases, the throughput is limited by the Iub/Iur transport network, especially in the initial deployment phase. As HSPA traffic is increasing in the network, most operators will expand their transport network to further enhance user experience.

9.2 Congestion Avoidance Mechanisms

Congestion occurs whenever the available link capacity is exceeded by the sum of the current flow rates. Congestion situations should be resolved in some way because they result in data loss. There are different means of controlling congestions. Congestion avoidance mechanisms can be classified based on the layer (e.g., data link routing or transport layer congestion control) where the mechanism operates, or on the duration of congestion. Note that solutions good for short-term congestion are not always good in the case of long-term overload, and vice versa. In most cases, a combination of mechanisms is used because overloads of various durations can occur in all networks [18].

Figure 9.6 depicts the congestion avoidance mechanisms proposed for use in the WCDMA RAN transport network. For very short data bursts, the most appropriate solution is to apply adequate buffers and QoS differentiation. This provides small delay for high-priority traffic, while buffering of low-priority, less delay-sensitive traffic increases utilization. For traffic with no fixed bandwidth requirement, flow control can be used to adapt the bit rate to the actually available resources. For guaranteed bit rate (GBR) services call admission control (CAC) can be used to check whether or not there is enough free capacity in the system. With adequate link dimensioning, the desired grade-of-service (GoS) can be guaranteed for a given traffic model.

Not all of these solutions are necessary in all cases; the optimal solution depends on several factors (e.g., on the cost of the transport network links and on the functionality available in the different equipments). For example, if the transport network links are very cheap, then they can be

[1] The bit rate available for ATM cells is 1920 kbps in the case of E1, 1536 kbps in the case of T1, and 33,920 kbps in the case of E3.

Congestion Duration	Congestion Control Mechanism
Long	Link dimensioning
↓	Call admission control
Short	Flow level congestion control
	Buffering and QoS differentation

Figure 9.6 Congestion duration versus congestion avoidance mechanism.

dimensioned according to the peak air interface capacity, and in this case probably no flow control or CAC is needed. However, in cases when the transport network links are costly, cooperating congestion control solutions on the different timescales can be used to optimally utilize the transport network links.

9.2.1 Buffering and QoS Differentiation

Buffering and QoS differentiation can improve packet-level performance. For example, we can give priority to the delay-sensitive packets or we can use larger buffers for best-effort packets. Both methods can improve system performance.

If we have more traffic classes with different QoS/GoS requirements, one option is to have completely separate (dedicated) resources for each traffic class. In this solution there is no interaction among traffic classes. As another option, we can share the resource among the traffic classes, but in this case we have to use proper scheduling and a buffering method that takes into account the QoS/GoS requirement of the traffic classes. In general, the resource-sharing method outperforms the application of separate resources and requires smaller link capacity.

Popular packet scheduling methods include [19]

- *Round Robin. This algorithm serves the queues in a cyclic order, ignoring the channel quality conditions. This method is outstanding in terms of simplicity and fairness of resource-sharing among queues.*
- *Priority. Always the queue with the higher priority is scheduled.*
- *Weighted Fair Queuing (WFQ). To each queue, a weight (w_i) is assigned. Queues are scheduled according their weights, so that user i achieves an average data rate of $\frac{Cw_i}{\sum_{j=1}^{N} w_j}$, where C denotes the link capacity.*
- *Rate shaping. Each queue has a given maximum transmission rate. If the incoming bit rate to the given queue is higher than the specified maximum, then the excess packets are dropped. This method is usually used in combination with other methods (e.g., priority scheduling).*

Popular buffering methods include

- Finite buffer with FIFO or LIFO dropping
- Random early detection dropping to inform TCP about congestion

CAC and the link dimensioning method must take into account applied buffering and QoS differentiation. The CAC method is developed for a given buffering and QoS differentiation method. Link dimensioning must take into account how the resources are shared among different flows and how well the flows with different QoS requirements are differentiated.

9.2.2 Flow-Level Congestion Control

The task of flow-level congestion control is to adapt the bit rate used by already ongoing flows to the available capacity. The used rate of GBR flows can be adapted to the available capacity, for example, in the case of adaptive multirate (AMR) voice codec or adaptive video streams. For Packet Switched (PS) traffic, the bit rate of the flow can be adapted more smoothly. TCP congestion control is a well-known example. In WCDMA RAN transport network HSPA, flow control is used to adapt the PS bit rate to the actually available capacity. HSPA flow control is detailed in "HSPA Flow Control" later in this chapter.

9.2.3 Call Admission Control

The task of Call Admission Control (CAC) is to ensure the desired QoS for traffic classes which have strict QoS requirements, usually GBR traffic. Whenever a new flow arrives, the CAC can decide to admit or reject it. A flow setup is rejected if the QoS requirements of the flows cannot be met, assuming that the arriving flow increases the traffic load. An already-established connection may also be released if it has lower priority than the arriving one in order to admit the new connection. The CAC can be based on calculation and/or measurements.

CAC algorithms based on analytical methods can vary in complexity. If the contribution of the GBR traffic to the total traffic load is high and detailed information about the flows is available, then the QoS of the flows can be evaluated using analytical calculations; and based on that, CAC decisions can be made. Such an algorithm is described in [20]. Otherwise, a simple admission control can be done, for example, reserving the peak or average bit rate of a GBR flow from the total link capacity. While a complex algorithm can allow high transport network utilization in a GBR-dominated case, a simple algorithm is often sufficient to reserve capacity because the traffic load of BE classes is high.

A CAC algorithm can also be based on measurement. In this case, the QoS is monitored and when the desired QoS is degraded, new flows are rejected and/or existing flows are released.

CAC can be end-to-end when it is executed only at the endpoints of the TNL flows. It can also be link-by-link when the TNL supports it, for example, in the case of AAL2/ATM-based transport network [16].

9.2.4 Link Dimensioning

Depending on traffic mix and the cost of the transport network capacity, different link dimensioning methods can be used. If transport network capacity is cheap, then *reasonable peak allocation* can be done. In this case, the total transport network capacity is configured to be high, such that the achievable air interface capacity is rarely higher than the dimensioning link capacity. On the other hand, if transport network capacity is costly and a traffic model is available, then *traffic model-based dimensioning* can be used. In case of higher-level aggregation, when the traffic of several Node Bs is aggregated, *overprovisioning* [21] can be used. In this case, the traffic of the transport network is continuously monitored, and the transport network links are upgraded as soon as the utilization reaches a limit.

Traffic model-based dimensioning can vary based on the properties of the dimensioned traffic. Traffic model-based dimensioning usually requires a target GoS. For a GBR service, such a GoS parameter can be, for example, the call rejection probability by the CAC; while for non-GBR services, for example, the experienced average throughput can be used. For GBR flows, the Kaufman-Roberts formula [22,23] can be used to determine the required capacity; while for PS traffic, an elastic calculation [24] can be used.

9.2.5 Congestion Avoidance Example

This section shows through an example how different congestion avoidance mechanisms cooperate.

We assign different traffic to different QoS levels, Figure 9.7 shows the used *buffering* and *QoS differentiation.* QoS level 0 and QoS level 1 have strict priority, whereas QoS level 2 and QoS level 3 share the remaining capacity based on the WFQ discipline. The vertical arrow in the figure indicates strict priority among the queues, QoS level 0 having the highest priority and the WFQ scheduler of QoS level 2 and QoS level 3 having the lowest priority.

Network synchronization and high-priority signaling belong to QoS level 0. Because of its highest priority, a short (e.g., 10 ms) buffer is sufficient for QoS level 0. As its volume is low compared to aggregated traffic demand, no *CAC* or *flow-level congestion control (FC)* is needed.

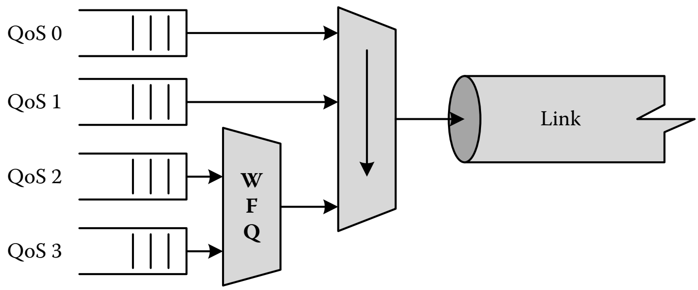

Figure 9.7 Example for buffering and QoS differentiation.

QoS level 1 is used for GBR services. It also has a short buffer, but is slightly longer (e.g., 20 ms). A simple CAC is used that reserves the peak rate of each flow end-to-end. The bandwidth reserved but not used can be reused by lower-priority traffic.

QoS level 2 is used for packet-switched interactive services, and QoS level 3 for packet-switched background services. To avoid starvation of background services caused by interactive ones. PS services can tolerate higher delay, therefore a larger buffer (e.g., 100 ms) can be used. This allows higher utilization of the bottleneck capacity. The peak rate of PS connections is usually high so no CAC is used for them. Instead, HSPA flow control is used to gracefully degrade the capacity used by PS flows.

For QoS level 0, the peak of network synchronization and high-priority signaling traffic can be reserved. However, the remaining capacity from this level can be used by lower-level traffic. For QoS level 1, the Kaufman-Roberts formula can be used to determine the required capacity. For PS traffic, elastic calculation can be used to determine the required capacity. The WFQ weights can be set by taking into account traffic volumes on the different QoS levels and also the fact that response time for interactive services is important. This can be done by, for example, multiplying the interactive traffic volume by a factor greater than 1.

9.3 HSPA Flow Control

Originally, HSDPA FC was intended to control radio scheduler queues in Node B (priority queue, PQ). On the one hand, these queues should be kept short enough to ensure that retransmitted data reaches its destination in a short time. On the other hand, these queues should be long enough to maximize air interface efficiency. In the case of HSDPA, the MAC-hs PQ in Node B can be long when the Uu interface is the bottleneck; in the case of HSUPA, there is no queuing in Node B, and so this is not an issue.

Later, the transport network became a potential bottleneck in the system because the increased air interface (Uu) capacity did not always come with

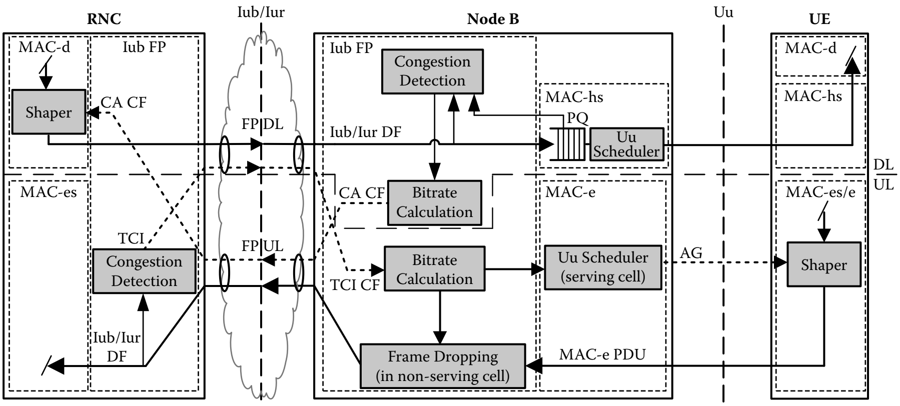

Figure 9.8 HSPA flow control architecture.

similarly increased transport network capacity in practice. Therefore, in addition to flow control, congestion control is also needed to avoid buffer overflow in the transport network. Thus, the algorithm used is often a combined flow and congestion control solution, which is often abbreviated as HSDPA/HSUPA flow control. In the transport network, it is the task of the flow control to provide high utilization, small data frame delay and loss. Flow control will share the TNL resources in a fair way among flows of the same priority.

Figure 9.8 depicts the HSPA flow control-related protocols and how the functionality is divided among the different protocol layers and nodes. The HSDPA part is shown in the upper part of Figure 9.8, whereas HSUPA is shown in the bottom part. In Table 9.1 the main differences between the mainstream (standardized) HSDPA and HSUPA flow controls are shown.

In the case of HSDPA, the Shaper in the RNC shapes the arriving MAC-d PDUs according to the signaled maximum flow bit rate. The Iub Framing

Table 9.1 Location of Different HSPA Flow Control Functionalities

	HSDPA FC	*HSUPA FC*
Congestion detection	Node B	RNC
Congestion indication	(to RNC)	To Node B in TCI CF
Bit rate calculation	Node B	Node B
Bit rate signaling	To RNC in CA CF	To UE in AG
Shaping	in RNC (MAC-d)	In UE (MAC-e)

Protocol (FP) puts them into Iub DFs and sends them to Node B. Each of these frames contains frame sequence number (FSN) and some of them contain the delay reference time (DRT). The Congestion Detection part in Node B aims at finding out the congestion level of the transport network, based on these pieces of information and on the length of the PQs. If transport network congestion is detected, the bit rate calculation part will be informed. The Bitrate Calculation part in Node B calculates the allowed maximum bit rate at which the RNC can send data to Node B to resolve the congestion situation. If the calculated bit rate of the flow changes, then the Shaper will be informed about the new bit rate through the capacity allocation (CA) control frame (CF).

In the case of HSUPA, the Shaper part is in the UE and Congestion Detection is in the RNC. The Shaper in the UE shapes the outgoing MAC-e PDUs according to the signaled maximum flow bit rate, and sends them to Node B. The MAC-e layer in Node B demultiplexes MAC-es PDUs and sends them to the Iub FP layer. The Iub FP in Node B puts arriving MAC-es PDUs to Iub DFs and forwards them to the RNC where Congestion Detection is performed based on DF fields similar to those of HSDPA. The Bitrate Calculation part is notified about congestion in the TNL Congestion Indication Control Frame (TCI CF). The Uu scheduler sends scheduling grants to the UE. There are two types of scheduling grants: absolute grant (AG) and relative grant (RG). AG is used to signal the maximum allowed bit rate by the transport network at which the UE may send data to Node B taking into account congestion information from the Congestion Detection part. The RG can modify this rate up/down in the serving cell, or only down in the non-serving cell. We assume that RG is not used for flow control purposes.

In the next section, "Congestion Detection and Network Monitoring," we discuss how *congestion detection and network monitoring* is performed. In the subsequent section, we discuss congestion action. ("Per-Flow HSPA Flow Control Solutions" and "Aggregated HSPA Flow Control Solutions") HSPA flow control algorithms can be implemented in different ways. Later sections describe *per-flow* and *aggregated* flow control. At the end, three case studies show examples for HSPA flow control.

9.3.1 Congestion Detection and Network Monitoring

Transport network congestion detection can be based on fields in Iub DFs or transport network-specific information. Flow control-related DFs and CFs are standardized in [7,12].

HS-DSCH Iub DFs contain the user data and transmit information about the amount of user data waiting in the RNC, called the user buffer size (UBS). They also contain information for congestion detection, the FSN, and the optional delay reference time (DRT).

The Iub/Iur Framing Protocol E-DCH DF contains the user data, the FSN, the connection frame number (CFN), and the subframe number. The CFN and subframe number are used for reordering, but can also be used to calculate the DRT. The FSN and DRT can be used for transport network congestion detection.

The DRT field, which can be found in the data frame payload, contains the value of a 16-bit reference counter in the sending node when the data frame is sent. The DRT counter is increased every 1 ms. The DRT can be used for dynamic delay measurements. In the case of HSDPA, the RNC does not send DRT in each frame. There is a flag to specify whether the frame contains DRT. In the case of HSUPA, each frame contains DRT information.

Apart from congestion detection based on DF fields, transport protocol-specific congestion detection techniques are also possible.

Transport network congestion detection for HSPA should be performed whenever a DF arrives at the receiving node. The following methods can be used:

- *Destroyed Frame Detection (DFD).* Due to the fact that ATM cells (not the data frames) could be lost in the ATM-based transport network, it is possible that only a part of a data frame is lost. When the segmented data frame is reassembled, it is possible to detect that a part of the data frame was lost.
- *FSN gap detection.* The 4-bit FSN in the data frame can be used to detect missing data frames (which were fully lost).
- *Dynamic Delay Detection (DDD).* In the case of HSDPA, the DRT field in the data frame contains the value of a reference counter in the RNC when the data frame was sent. The DRT is compared to a similar reference counter in Node B when the data frame is received. In the case of HSUPA, the Node B DRT is compared to a similar reference counter in RNC when the data frame is received. The difference between the two counters increases when the transport network buffer is built up. Congestion is detected when this difference increases too much compared to the minimum difference.
- *Other transport network protocol-specific methods.* For example, Explicit Congestion Notification (ECN) field [25] is used in IP header for lossless TCP congestion control but can also be used for HSPA transport network congestion control.
- *Direct TNL information.* For example, TNL buffer length can be used directly, but this makes congestion detection dependent on the transport network and requires communication between the TNL and the congestion control.
- *Data frame rate measurement.* This method can be used to determine the severity of the congestion and to get information about the state of the transport network.

Non-congestion-related data frame loss in the transport network can result in false congestion detection; therefore, it should be avoided. If it is not possible to keep it very small, then one might consider adding filtering for the loss-based congestion detection, depending on the used TNL technology and typical loss patterns. However, this is not needed in the case of the common TNL technologies used.

Dynamic Delay can be differentiated by its severity by defining different dynamic delay limits. These DDD limits must be configured by taking into account the frame delay variation caused by higher-priority traffic. The limits must be set higher than the non-congestion-related frame delay variation; otherwise, congestion will be detected even when there is no congestion in the transport network, and these false congestion detections will result in performance degradation. In the case of very small transport network buffers or very high frame delay variation caused by other traffic, it is recommended not to use DDD at all.

Different congestion events can be classified and different actions can be taken. For example, dynamic delay exceeding a small limit can be classified as *soft congestion* and all other congestion events (DFD, FSN, and DDD for large limit) can be classified as *hard congestion*.

HSDPA and HSUPA congestion detection are based on similar principles but in the case of HSUPA, congestion detection and bit rate calculation are separated. HSUPA congestion detection takes place in the RNC, whereas the bit rate calculation takes place in Node B. Therefore, congestion indication is not used in the case of HSDPA. In the case of HSUPA, the detected congestion and its severity are reported to Node B by a TCI CF, if no TCI CF was sent for a given time. The TCI contains a congestion status field, which can indicate no congestion, congestion due to delay build-up, or congestion due to frame loss.

9.3.2 Congestion Action

Congestion action refers to all the actions needed to resolve congestion, that is, bit rate calculation, notification, and shaping. The bit rate calculation algorithm itself is not standardized; each vendor can implement its own solution. Shaping and notification are standardized.

While the purpose of HSDPA flow control [17] and HSUPA flow control is similar, there are significant differences. First, for HSUPA, only the transport network bottleneck must be regulated, while for HSDPA there are also Uu scheduler queues in Node B to be regulated (called MAC-hs PQs [17]). This means that HSDPA FC must deal with Uu and transport network bottleneck; but in case of HSUPA FC, the Uu bottleneck is completely handled by the Uu scheduler. Second, HSUPA can be in soft handover, while HSDPA cannot. For HSUPA, this means that for the same radio bearer, there can be several (one serving and zero or more non-serving) flows to be controlled. Third,

in the case of HSDPA, the FC has a more direct impact on flow rates than Iub. For HSDPA, the explicit rate should be given, whereas for HSUPA it can be given but other ways are also possible.

For HSDPA, based on UBS, Uu-related radio scheduler queue information and the output of congestion detection the FC algorithm in Node B decide how many MAC-d PDUs can be transmitted from the RNC for a given flow. This is reported to the RNC using the HS-DSCH CA CF. The CA CF defines maximum MAC-d PDU *length*, HS-DSCH *credits*, and HS-DSCH *interval* and HS-DSCH *repetition period* values. The MAC-d shaping in RNC ensures that within a given interval, not more PDUs are sent than specified in the *credits* field. Repetition period defines how many times the interval and the credits are repeated. A newly received CA CF overrides the old one. Setting the repetition period unlimited allows one to define an allowed shaping rate; e.g., max. PDU length 42 octets, credits 4, and interval 10 ms settings define a shaping rate of 134.4 kbps. In the case of flexible RLC, instead of checking the number of PDUs sent within an interval, the number of octets is limited per interval to avoid bit rate loss due to smaller RLC PDUs (see details in “HSPA Framing Protocol”).

The CA CF can be used in a rate- or credit-based manner for HSDPA. The main advantage of rate-based usage is that it generates moderate CA load. High CA load should be avoided because it requires high processing power from the nodes. There are practical problems with using CA in a credit-based manner. It requires more intense signaling; moreover, CAs can be lost. Therefore, we do not know by when all the credits will have been used. A window-based solution would solve this problem but an acknowledgment window is not supported by the standard. Therefore, we discuss rate-based CA in this chapter.

The HSUPA air interface scheduler (Uu scheduler) operates by sending scheduling grants to UE and receiving scheduling requests from UE [3]. Only the scheduling framework is standardized, the scheduling algorithm itself is not. There are two types of scheduling grants: AG and RG. AGs can be sent only by the serving cell and transmitted over the E-DCH Absolute Grant Channel (E-AGCH), which is a shared resource among all users of the cell. The AG defines how many bits can be transmitted every TTI and thus a maximum limit of the data rate. The AG is valid until a new scheduling grant is received. The RG can modify this rate up/down in the serving cell, or only down in the non-serving cell. The UE indicates, by a flag called Happy Bit, whether or not it would benefit from a higher rate grant.

In the case of HSUPA, until the first rate reduce request is received, the Uu scheduler behavior is not affected by FC. Based on air interface conditions, hardware resources, and Happy Bit information, the Uu scheduler decides the granted bit rate represented by the AG. Upon receiving a rate reduce request, the scheduler decreases the granted bit rate.

HSUPA FC is designed to provide fair throughput sharing among the flows sharing the same transport network bottleneck when the transport network is limiting the throughput. The behavior of flows is regulated by the Uu scheduler until a transport network congestion is detected. The reason for this is that as long as the transport network is not a bottleneck, it is the task of the Uu scheduler to utilize the air interface as much as possible and to provide fairness among the flows. The Uu scheduler increases the granted bit rate with reasonable speed to avoid large interference peaks. This also ensures that sudden overload of the transport network is avoided.

9.3.3 Per-Flow HSPA Flow Control Solutions

For per-flow flow control, congestion detection and bit rate calculation are done independently for all flows. This approach is flexible for different transport network topologies, solutions, and configurations, that is, regardless of how the transport network bottlenecks are distributed. Consquently, per-flow solutions need less maintenance with future updates and changes of the WRAN system.

Thus, a per-flow solution supports different transport network bottlenecks for flows of the same Node B and TNL QoS differentiation among the flows. Some flows of a Node B might experience different bottlenecks; for example, when some flows are transmitted over Iur links, others are not; or when some flows can be routed on different paths on the Iub interface.

Per-flow solutions may partly rely on some aggregated information, as in [26]. In that reference, the flow control itself works per flow, but the initial rate for each flow is determined based on Node B-level aggregated information.

To achieve fair bandwidth sharing among flows sharing the same bottleneck, Additive Increase/Multiplicative Decrease (AIMD) can be used. Other congestion control styles such as Additive Increase/Additive Decrease (AIAD), Multiplicative Increase/Multiplicative Decrease (MIMD), and Multiplicative Increase Additive Decrease (MIAD) have also been studied in the literature. However, assuming synchronous congestion signals and static bandwidth, AIMD proved the only fair and stable choice among them [27]. In [27], it is shown that AIMD guarantees convergence to fairness; all flows converge to an equal share of resources in steady state, where no flows join or leave.

9.3.4 Aggregated HSPA Flow Control Solutions

Aggregated flow control solutions aim to share the available bandwidth among the flows based mainly on aggregated network information. Aggregation may be performed on multiple levels, for example, per cell or per Node B. Per-flow parts may also be present in such an algorithm

but the dominant part is performed based on aggregated information. Aggregated solutions have an overall picture of the system; that is why they can react faster to changes in some cases and provide a fairer distribution of resources.

An aggregated solution requires detailed information about the transport network bottleneck(s) and QoS solution; it also should support aggregated transport network connections, where flows of several Node Bs can experience bottleneck. While such a solution is not impossible, its complexity is too high compared to the achievable gains. The aggregated solution requires continuous maintenance when the transport network topology or the transport network QoS solution used changes.

It is possible to have an aggregated solution that does not have full information about the transport network. In this case, however, many of the above advantages disappear.

9.3.5 Case Study: Aggregated HSDPA Flow Control Solution

In [17] an aggregated HSDPA flow control algorithm is proposed. Apart from controlling the MAC-hs PQ, it also solves the congestion situation on the transport network. The algorithm is a rate-based FC solution, which includes a per-flow part, and per-cell and per-Node B-level aggregated information. The per-flow part is responsible for the fast reaction to a congestion situation in the transport network or long PQ delays. The cell-level aggregation estimates the frequency of scheduling a PQ in the given cell. This is used for estimating the air interface bit rate of the PQ. The Node B (or transport network) level aggregation approximates the available transport network capacity for HSDPA and distributes it among the flows. The notations introduced in this section reflect the different aggregation levels and the types of variables. Variables without superscripts denote per-flow variables, while superscript *cell* denotes cell level and superscript *Node B* means Node B-level aggregation.

In the case of large transport network buffers, DFD and FSN cannot provide efficient congestion detection because protocol problems occur before the transport network buffer becomes full. DDD provides congestion detection in this case, and it also decreases transport network delay and loss in the case of small transport network buffers. The shaping rate is calculated separately for every active flow, once per 100 ms. The 100-ms value is a compromise between fast reaction, low CA frequency, and low calculation complexity. The time of this calculation, every 100 ms, is called the tick time, which is not synchronized for the different flows. The FC distributes the estimated transport network capacity among the flows proportional to their estimated available Uu rate. Thus, a newly arriving flow gets its share of the transport network capacity, while the rate of ongoing flows decreases.

The short-term transport network congestions are taken into account by a coefficient. The calculated rate is translated to CA format and sent to the RNC. The estimated transport network link capacity ($BW_{estTN}^{Node\ B}$) and the sum of estimated Uu capacities (BW_{estUu}^{NodeB}) are provided by Node B-level aggregation. The estimated Uu rate (BW_{estUu}) is provided by the cell-level aggregation. The FC calculates the shaping rate as follows:

$$BW_{calc} = \min\left(\frac{BW_{estUu}}{BW_{estUu}^{NodeB}} \cdot BW_{estTN}^{NodeB}, BW_{estUu}^{NodeB}\right) \cdot Q_{Iub}, \tag{9.1}$$

where Q_{Iub} is a coefficient that reacts fast on a detected congestion. Q_{Iub} is set to 0.5 if transport network congestion has been detected since the last tick; otherwise, it is increased by 0.05 at each tick. The maximum value for Q_{Iub} is 1.

9.3.5.1 Node B-Level Aggregation

The purpose of Node B-level aggregation is to estimate the available transport network capacity. The value of BW_{estTN}^{NodeB} is updated once per second. It is based on the number of transport network congestions detected by the per-flow ticks of the different flows during the last second, called $N_{IubCong}^{NodeB}$. As there is one tick in every 100 ms per flow, one flow can contribute, at most, ten flags during the evaluation period.

BW_{estTN}^{NodeB} is limited by a preconfigured minimum and maximum rate (called BW_{minHS}^{NodeB} and BW_{maxHS}^{NodeB}, respectively), which must be configured according to the transport network configuration. If $N_{IubCong}^{NodeB}$ is greater than or equal to a predefined threshold (e.g., 5) or the number of active flows (N_{flow}^{NodeB}), then the $BW_{estTN}^{Node\ B}$ is decreased by 2% of BW_{maxHS}^{NodeB}. If the $BW_{estTN}^{Node\ B}$ was not decreased in the last 10 seconds, then it is increased by 1% of R_{maxHS}^{NodeB} every second. The constants are determined by considering how frequently and to what extent the transport network capacity typically changes.

BW_{estUu}^{NodeB} equals the sum of BW_{estUu} for all flows in Node B. This variable is used for distribution of R_{estTN}^{NodeB} among the flows.

9.3.5.2 Uu Rate Estimation

To predict the Uu rate of a flow (BW_{estUu}), the possible peak Uu rate of the flow (BW_{peakUu}) is divided by the average number of competing flows (N_{flow}^{cell}). For long PQs, the estimated values are further reduced. The BW_{peakUu} is reported by the radio scheduler.

$$BW_{estUu} = Q_{t_{PQ}} \cdot \frac{BW_{peakUu}}{N_{flow}^{cell}} \tag{9.2}$$

where $Q_{t_{PQ}}$ is a coefficient calculated from the estimated time to serve all PDUs in the PQ (t_{PQ}). The time t_{PQ} is calculated as follows:

$$t_{PQ} = \frac{b_{PQ}}{\frac{BW_{peakUu}}{N_{flow}^{cell}}} \tag{9.3}$$

where b_{PQ} is the length of the PQ (in bits). The value of $Q_{t_{PQ}}$ is set to keep the delay at an appropriate level. Its value is 1 if t_{PQ} is smaller than 50 ms; it is 0 if t_{PQ} is larger than 150 ms and decreases linearly if t_{PQ} is between 50 ms and 150 ms.

9.3.6 Case Study: Per-Flow HSUPA Flow Control Solution

In [28,29], a rate-based per-flow HSUPA transport network congestion control solution is presented.

When transport network congestion is detected, FC dominates the behavior. During this time, flows are regulated according to an algorithm, which conforms with the AIMD property. Multiplication with a coefficient provides the multiplicative decrease, and a constant increase in rate after reduction provides the additive increase property. The AIMD property is met only for the serving cell behavior. However, a MAC-e PDU is normally received in the serving cell with a higher probability; thus end-user fairness is dominated by serving-cell behavior.

9.3.6.1 Transport Network Congestion Detection

The transport network congestion detection part of the algorithm is performed whenever a data frame arrives at the RNC. Two different congestion detection methods are used: FSN gap detection and DDD.

9.3.6.2 Bit Rate Calculation

Whenever a TCI is received by Node B, it triggers a *congestion action* by the Flow Control entity. Depending on the severity of the congestion, a reduce request with a certain coefficient is issued. Different coefficients are applied in the case of soft and hard congestions. A different coefficient can also be used for the first TCI received for a flow. The motivation for this is that when there is no TNL congestion at all, the Uu scheduler increases the granted bit rate with a higher speed. Consequently, these UEs can potentially overload the transport network more so than UEs already limited by the effect of flow control.

Depending on whether it is a serving-cell flow or a non-serving-cell flow, the rate reduce request is issued to the Uu scheduler or to the Frame dropping functionality.

9.3.6.3 Congestion Action in the Serving Cell

Until the first rate reduce request is received, the Uu scheduler behavior is not affected by Flow Control at all. Based on air interface conditions, hardware resources, and Happy Bit information, the Uu scheduler determines the granted bit rate represented by the AG.

Upon receiving a rate reduce request, the scheduler decreases the granted bit rate by sending a new AG. Additionally, when a rate reduce request is issued for a flow, the scheduler is not increasing the absolute grant of that flow with more than a predefined rate (e.g., 20–200 kbps/s). The value is determined based on typical transport network bit rate and the typical number of parallel flows.

The Uu scheduler maintains an allowed bit rate according to the above algorithm, and the bit rate represented by the sent AG must be lower than this allowed bit rate. Note that there are only a certain number of different possible AG values to send. Consequently, the reduction in allowed bit rate and the reduction in AG may be different, according to the granularity of the possible AG values.

9.3.6.4 Congestion Action in the Non-Serving Cell

A TCI received in the non-serving cell will not trigger rate reduction by RG because a MAC-e PDU is received in the best cell (usually the serving cell) with a higher probability. Consequently, if we reduced the bit rate due to transport network limitations in the non-serving cell, we might reduce the bit rate of the end user unnecessarily. However, congestion action still needs to be taken; thus, a fraction of the received MAC-es PDUs are dropped. If these PDUs are not received in the serving cell, then RLC AM still retransmits these missing PDUs.

A *forwarding coefficient* determines the probability that a received MAC-e PDU is forwarded. It is 100% at initialization, and each received reduce request decreases it. It is gradually increased to 100% afterward. Note that this behavior does not conform to AIMD, but a MAC-e PDU is normally received in the serving cell with a higher probability; thus, end-user fairness is dominated by serving-cell behavior.

9.3.7 Other Flow Control Solutions

In [30], the authors introduce *cross-layer backpressure* in the RNC, which allows good transport network utilization when the transport network bottleneck buffer is in the RNC. Using CA Credits calculation the algorithm not only takes into account the information provided by Node B, but also the state of the AAL2/ATM level queues in the RNC. Thus, they assume that the Iub transport network consists of one link, and the state of the AAL2 buffer at its input is known. In this way they can perform a pro-active,

aggregated congestion control. The scope of the algorithm is to control the load of the low-priority AAL2 buffer so that the delay on the AAL2 layer is not exceeding the maximum allowed delay.

9.4 HSPA Framing Protocol

The Iub/Iur Framing Protocol defines the DFs and CFs to be used over the WCDMA RAN transport network to transmit user data and transport network layer control information.

For HSDPA, the DFs and CFs are defined in [7]. Two types of HS-DSCH Frame Protocols exist for the HS-DSCH data transfer procedure: (1) HS-DSCH Frame Protocol Type 1, including HS-DSCH Data Frame Type 1 (Figure 9.9) and HS-DSCH Capacity Allocation Type 1 Control Frame (Figure 9.10); and HS-DSCH Frame Protocol Type 2, including HS-DSCH Data Frame Type 2 (Figure 9.11) and HS-DSCH Capacity Allocation Type 2 Control Frame (Figure 9.12). Type 1 (often described without explicitly stating the type) is the original solution. FP Type 2 was introduced to support higher bit rates; it supports flexible RLC, higher PDU sizes, octet aligned PDUs, and multiplexing of several logical channels. It also decreases transport network overhead compared to Type 1. HS-DSCH Frame Protocol Type 2 is selected if the HS-DSCH MAC-d PDU Size Format Information Element (IE) in NBAP (Node B Application Part) is present and set to “Flexible MAC-d PDU Size.”

HS-DSCH Capacity Allocation Control Frame (CA CF) is used by Node B to control the user data flow from the RNC. CA CF informs the RNC about *TN Congestion Status*, about the priority of the flow (*CmCH-Pi*), and about the *maximum PDU length* allowed. It defines the amount of PDUs (*Credits*) that can be sent by the RNC. The *Interval* indicates the time interval during which the Credits are granted. By setting the *Repetition Period* to a high value, *Credits* are regenerated periodically; the length of the period is

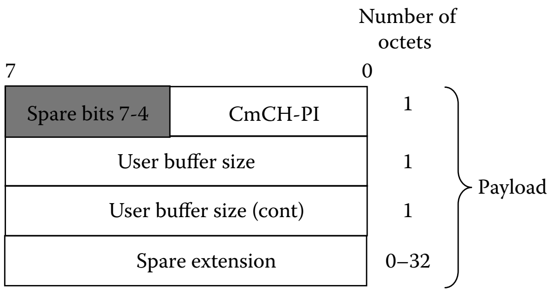

Figure 9.9 HS-DSCH data frame Type 1 structure. (*Source*: From 3GPP, TS 25.435 V7.1.0, UMTS UTRAN Iub Interface User Plane Protocols for Common Transport Channel Data Streams.)

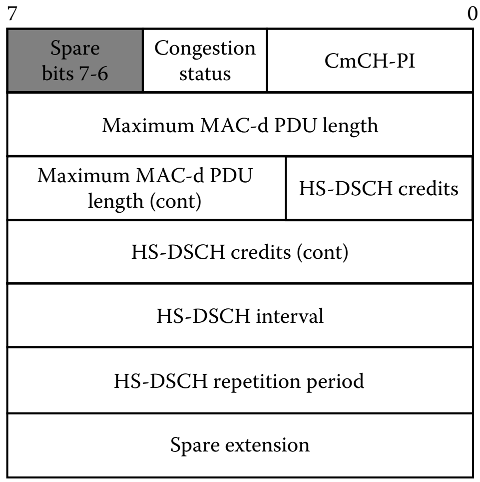

Figure 9.10 Capacity allocation control frame Type 1 payload structure. (*Source*: From 3GPP, TS 25.435 V7.1.0, UMTS UTRAN Iub Interface User Plane Protocols for Common Transport Channel Data Streams.)

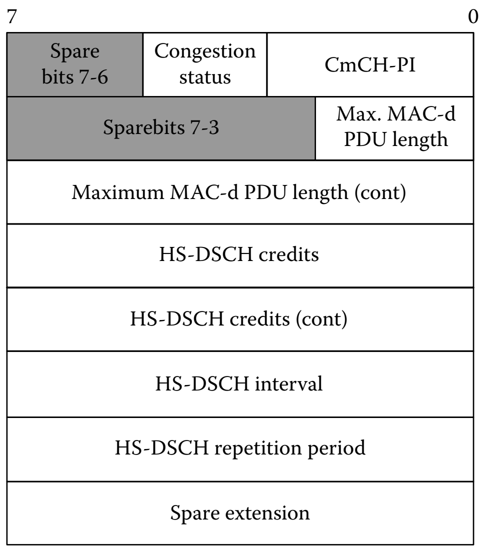

Figure 9.11 HS-DSCH data frame Type 2 structure. (*Source*: From 3GPP, TS 25.435 V7.1.0, UMTS UTRAN Iub Interface User Plane Protocols for Common Transport Channel Data Streams.)

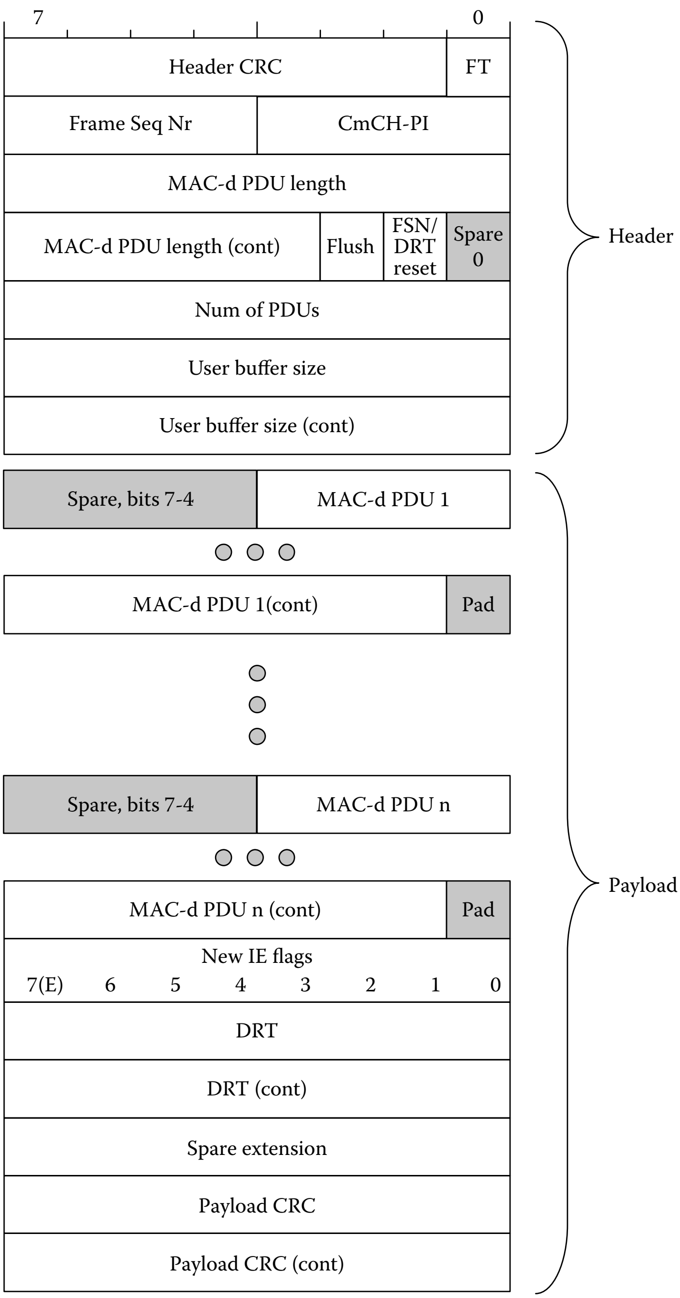

Figure 9.12 Capacity allocation control frame Type 2 payload structure. (*Source*: From 3GPP, TS 25.435 V7.1.0, UMTS UTRAN Iub Interface User Plane Protocols for Common Transport Channel Data Streams.)

defined by the *Interval*. The *Repetition Period* can be set to infinite. In this case, the CA CF represents a shaping bit rate because, in every Interval, the defined amount of PDUs can be sent until a new CA CF is received by the RNC. Special values can indicate zero or unlimited allocation.

A Type 2 CF allows longer PDU length (up to 1504 octets) than Type 1 because it has a longer *Maximum PDU length* field, and this length is defined in octets instead of bits. However, it requires octet-aligned PDUs, while Type 1 does not. A Type 2 CF supports flexible PDU sizes because the scheduling is defined in octets per interval instead of PDUs per interval. The CF fields are similar; but while for Type 1, the actual number of PDUs is restricted in each interval, for Type 2 the amount of octets is restricted. The number of octets is calculated in the shaper as the Maximum PDU length times the Credits. This, for example, allows sending two times *Credits* PDUs, if the actual PDU size is half of the maximum. This requires a more complex shaping algorithm, which was necessary to make the bit rate represented by the shaping independent of the actual PDU length in the case of flexible PDU size.

The HS-DSCH DF transfers PDUs between the Node B and RNC entities. The DF fields indicate the *FSN*, the priority of the flow *CmCH-Pi*, the *User Buffer Size* (UBS), and optionally the *Delay Reference Time (DRT)*. It contains the transmitted PDUs and information about their number and length. CRC provides error checking functionality for header and payload, and Spare Extension allows extending the frame in the future in a backward-compatible way.

The Type 1 DF allow only PDUs of the same size to be sent in the same DF. It defines PDU size in bits; and for octet-aligned PDUs, it has 8-bit *Spare* and *Pad* bits in total. The Type 2 data frame requires octet-aligned PDUs but allows PDUs of different size and logical channel to be sent in the same data frame. No extra bits per PDU are required, as in the case of Type 1. A Type 2 DF became necessary to allow higher PDU size and different-sized PDUs in the same data frame. By allowing different PDU sizes in a single data frame, the total overhead became smaller. (With flexible RLC, PDUs can be smaller than the maximum size to eliminate padding and the need for multiplexing parts of several SDUs to the same PDU.) The frame structure in Figure 9.11 is simplified; only the fields relevant for FDD and dedicated transmission are shown.

The HS-DSCH Capacity Request control frame (Figure 9.13) can be optionally sent by the RNC to Node B to indicate the *User Buffer Size* and request the sending of CA CF. As the data frame also includes this information, the use of this control frame is optional.

For E-DCH, the data and control frames are defined in [12].

The RNC uses the TNL Congestion Indication control frame (TCI CF, Figure 9.14) to signal detected transport network congestion to Node B. It informs Node B about *Congestion Status*, which can be no congestion,

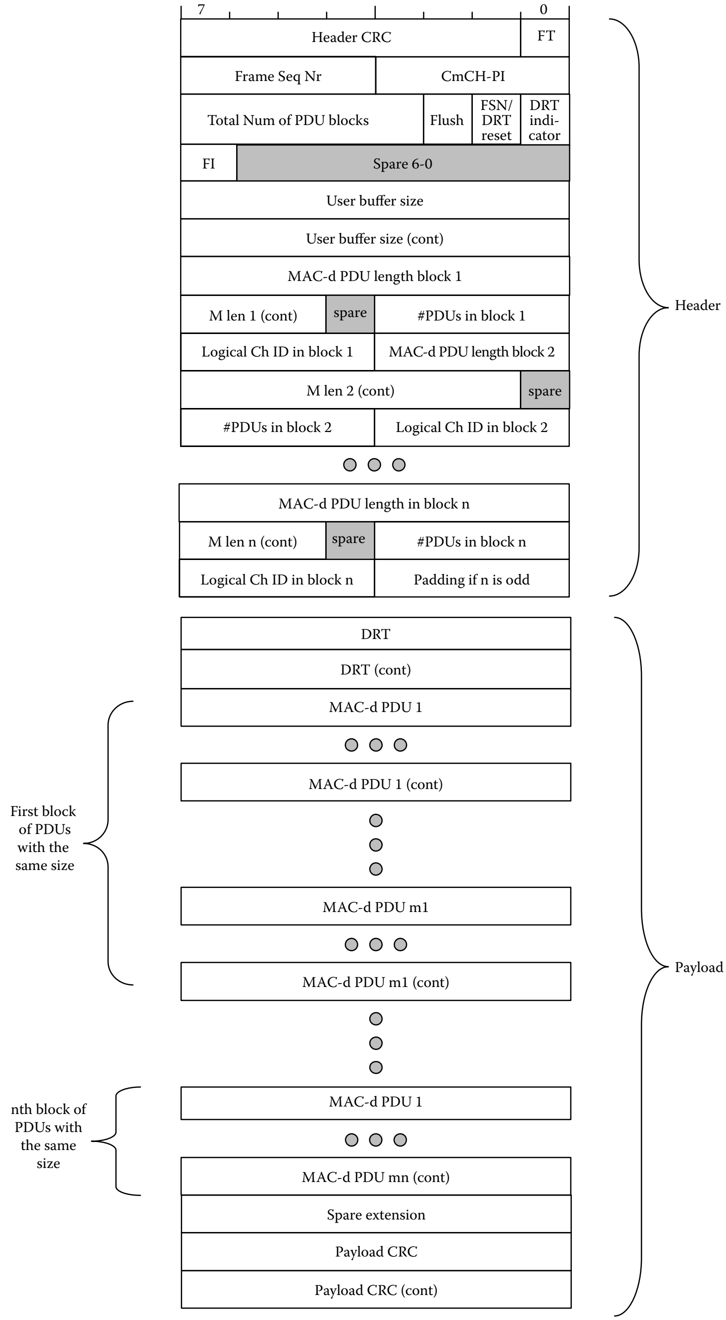

Figure 9.13 HS-DSCH Capacity Request payload structure. (Source: From 3GPP, TS 25.435 V7.1.0, UMTS UTRAN Iub Interface User Plane Protocols for Common Transport Channel Data Streams.)

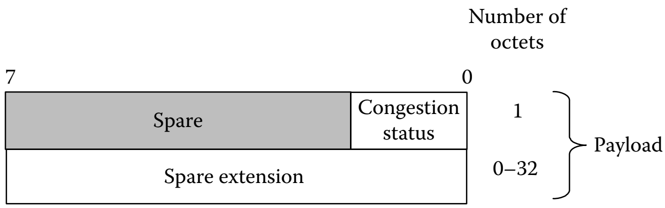

Figure 9.14 Structure of the TNL congestion indication (TCI) control frame. (*Source*: From 3G PP, TS 25. 427 V 7.5.0, UTRAN Iub/Iur Interface user Plane Protocol for DCH Data Streams (Release 7). October 2007.)

congestion due to loss or due to delay build-up. Upon receiving a TCI CF indicating congestion, Node B should reduce the bit rate of that user.

The E-DCH UL data frame (Figure 9.15) transfers MAC-es PDUs from the Node Bs to the RNC. Its fields indicate FSN Connection Frame Number (CFN), and Subframe Number. The CFN indicates the time when the data was received. Several subframes can be multiplexed to the same data frame, each having its separate header information. More than one subframe can be used in case of 2-ms TTI, when the data from up to five TTIs can be multiplexed in a single data frame. For each subframe, the Number of HARQ Retransmissions, the Subframe Number, the Data Description Indicator (DDI), and the Number of MAC-d PDUs (N) are indicated. The CFN and the last Subframe Number together can be used to measure dynamic delay; thus, no direct indication of *DRT* is needed. The data frame contains the MAC-es PDUs received in Node B. In the simplest case, the data frame contains a single subframe and a single MAC-es PDU. Cyclic Redundancy Check (CRC) provides error checking functionality for header and payload, and Spare Extension allows extending the frame in the future in a backward-compatible way.

9.5 TNL Overhead

We define the transport network overhead as the number of octets needed to be transmitted over the transport network, divided by the transmitted user-level IP octets within. It depends on the size of the data frame and the used transport network protocols. Apart from the transport protocol overhead, the overhead value also contains the Iub/Iur Framing Protocol and the RLC overhead. For E-DCH, the MAC-es overhead is also considered.

For the UDP/IP/Ethernet solution, the overhead depends much more on the data frame size, as in the case of AAL2/ATM. This is because for AAL2/ATM, most headers are on segmented PDUs, resulting in a fixed percentage, while for UDP/IP/Ethernet the headers are large, but apply to the data frame only once. If MLPPP is used as L2 for UDP/IP, then IP

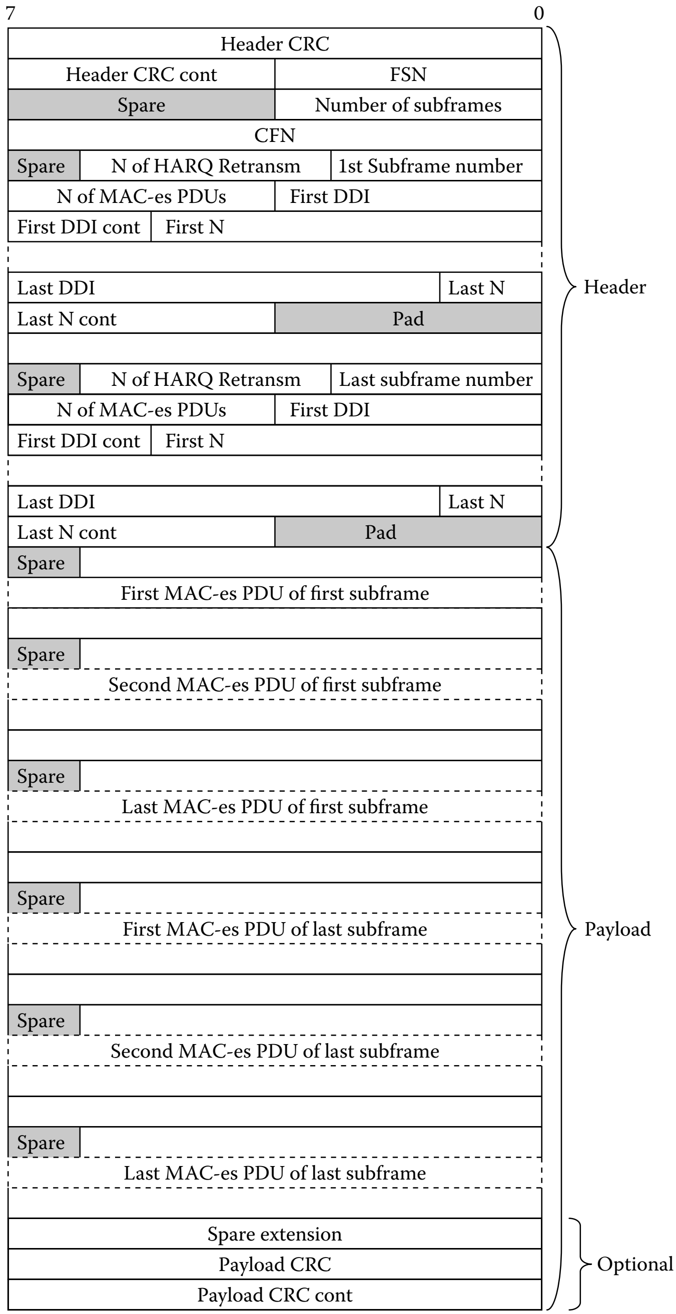

Figure 9.15 E-DCH UL data frame structure. (*Source*: From 3G PP, TS 25. 427 V 7.5.0, UTRAN Iub/Iur Interface user Plane Protocol for DCH Data Streams (Release 7). October 2007.)

Table 9.2 HSDPA Transport Network Overhead Examples

RLC PDU Size (octets)	*DF Type*	*PDUs per DF*	*ATM OH*	*IP OH*
42	1	10	1.32	1.27
82	1	5	1.28	1.24
302	2	2	1.23	1.14

header compression becomes possible and overhead in the case of small throughput is significantly reduced.

In the case of HSDPA, the transport network overhead primarily depends on the number of MAC-d PDUs carried in the HS-DSCH data frame. User data is segmented into RLC PDUs. For user data, the MAC-d layer does not add any headers so the RLC PDU is identical to the MAC-d PDU. RLC PDU size can take on one of the following fixed values: 42 or 82 octets, or it can be of flexible size.

Table 9.2 provides examples for HSDPA transport network overhead. The size of the data frame depends on the RLC PDU size and on the typical number of PDUs in a data frame. This is determined by the CA calculation and the CA shaping, and that is independent from air interface scheduling. For this example calculation, we assumed a given typical number of PDUs per data frame. Also, for the flexible PDU case, we assumed a maximum size of 302 octets and that all PDUs in the data frame are of the same

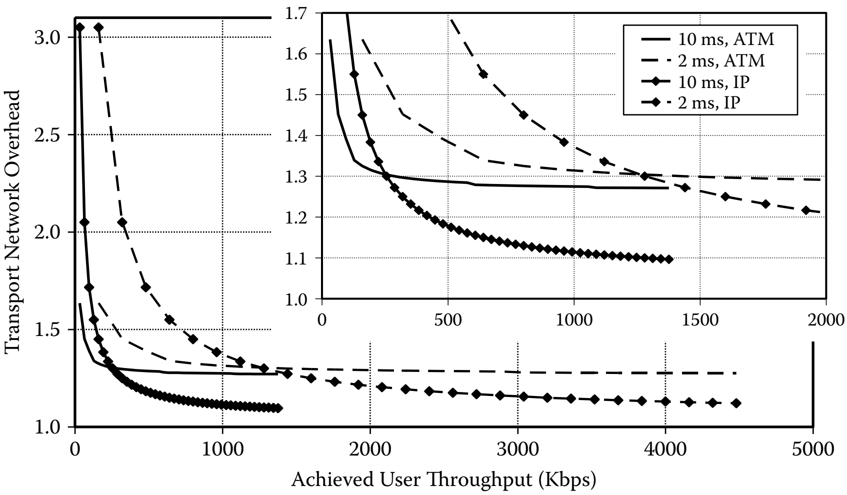

Figure 9.16 HSUPA transport network overhead as a function of the achieved user throughput.

maximum size. Because DRT use is not mandatory in all data frames, we assumed that the DRT is not present in the data frames.

In the case of HSUPA, the data frame size depends mainly on the achieved user throughput and on the TTI. This is because whenever a MAC-es frame is received, it is put into a data frame and transmitted over the transport network. For 2-ms TTI, the MAC-es PDUs from one or more TTIs may be bundled into one data frame before being transferred [12]. With bundling up to five PDUs, the Transport Network overhead in the case of 2-ms TTI can be decreased very close to the overhead in the case of 10-ms TTI. In the examples in this section, no bundling was assumed for 2-ms TTI. Figure 9.16 depicts the overhead in the case of AAL2/ATM and UDP/IP/Ethernet transport network for both 10-ms and 2-ms TTI. There is very large overhead in the case of small throughput and UDP/IP/Ethernet transport because the data frame size is very small for small bit rates. The aforementioned MLLPP with header compression can decrease this overhead if it becomes necessary.

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